International Journal of Computer Science Engineering
and Information Technology Research (IJCSEITR)
ISSN(P): 2249-6831; ISSN(E): 2249-7943
Vol. 4, Issue 5, Oct 2014, 83-90

TCP CONGESTION CONTROL PROTOCOLS OVER UMTS WCDMA NETWORK

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ABSTRACT

Universal Mobile Telecommunication Network (UMTS) deal with various packet losses. These packet losses developed in network due to channel fading and shadowing. Contention in network should maintain by TCP end-to-end semantics and dependency on intermediate nodes is minimized. Recent development in 3G UMTS networks new services provide to users makes it necessary for improving TCP's efficiency and resource utilization. In this paper, we implement various TCP protocol's flavous on UMTS networks. TCP is reliable protocol but with a problem of slow start. This paper simulate the effiency of different TCP variants i.e. TCP RENO, TCP TAHOE, TCP SACK and TCP NEWRENO. Performance comparision is done between all of these TCP variants on UMTS network. Simulation results shows that which TCP version perform better in UMTS network and how long take time to send and receive a FTP file. These protocols are inspected based on 'FTP traffic sent (packets/seconds)', 'FTP traffic received (packets/seconds)' and TCP connection (Congestion Window)'.

KEYWORDS: Universal Mobile Telecommunication System (UMTS), Transmission Control Protocol Tahoe (TCP Tahoe), TCP RENO, TCP SACK. TCP NEW RENO & Operational Network Evaluation Tool (OPNET)

INTRODUCTION

Recent advances in Mobile technology perform changes the way of the communication of people. As people need to send various type of multimedia data the fast growth of wireless packet-switched networks is emerged, people send the data through the new 3G networks rather than the old 2G networks public Switched Telephone Network (PSTN) The new 3G technology has become a better option in terms of cost for users and various service providers, leading to massive growth of voice applications and video applications over IP networks. Transmission of multimedia content over wireless access networks (in particular, Third Generation Universal Mobile Telecommunication System (3G UMTS)) is growing exponentially and get popularity, and is predicted to develop new revenue streams for mobile network operators. This paper provides a simulation methodology for performing various flavors of Transmission Control Protocols (TCP) in terms of congestion in a UMTS network environment. This paper presents scenarios that has been developed with OPNET is used to evaluate the behavior of different Transmission Control Protocol (TCP) flavors in UMTS (Universal Mobile Telecommunications System) for 3rdGeneration (3G) mobile communications. The simulator includes various simulation models for the UMTS network environment which contains User Equipment, Base Station and Radio Network Control nodes. The functionalities that are implemented are those that deal with various TCP connection functions (i. e. to control the congestion) and traffic generation models. Several performance statistics scalar types (TCP sent segment sequence number and TCP congestion widow) are obtained.

UMTS PROTOCOL OVERVIEW

Figure 1 presents the basic architecture of the Universal Mobile Telecommunication System (UMTS) [7]. UMTS network is composed by the Core Network (CN) and the UMTS Terrestrial Radio Access Network (UTRAN). The UTRAN is composed by several Radio Network Subsystems (RNS) each one including a Radio Network Controller (RNC) that is connected with several base stations (node Bs) by Iub interfaces and Finally, User Equipments (UE) can be connected to one or more base stations (node Bs). The RNC performs, all the functions that are related with the allocation of radio resources i.e. bandwidth , RRM and QoS management while node Bs are responsible of performing physical layer functions such as synchronization, channel estimation or power control.

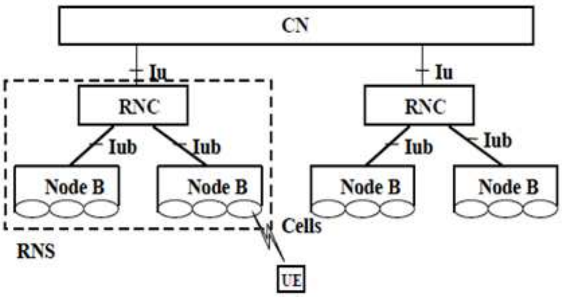

Figure 1: UMTS Architecture

For each multimedia service, characteristics and functionality must be set up from the source to the destination, maybe including not only the UMTS network [8] but also external networks. UMTS provides data rate up to 2Mbps for different QOS classes. Various QOS classes used in UMTS are:

- **Conversational Class:** Used for voice/video telephony data, this class has low end-to-end delay and low jitter. Communication is one way.
- **Streaming Class:** Used for streaming video data. This class has low jitter. Communication is one-way
- **Interactive Class:** Used for web browsing. With minimum loss, two-way
- **Background Class (Best Effort):** Used for email and background download, with low loss/error rate, one- way.

RELATED WORK

Probably the most challenging area will be the TCP congestion control [1] [2] in wireless networks as most current TCP implementations are designed for wired networks that rely on packet loss being a good indicator for network congestion. A congested network element is indeed a likely reason for a packet loss in wired networks with stationary hosts, but not in wireless networks. The motivation behind to use TCP in UMTS network was to add reliability on top of an unreliable IP network. The original TCP used a sliding window mechanism to control the transmission of data on TCP [6], TCP provide packet acknowledgements and segment sequence numbers which guaranteed a reliable data transmission and flow control on networks. Its congestion control, originated from wired networks, where congestion is the main reason for packet loss, comes under pressure in wireless networks. Because these networks are characterized by dynamically variable channel conditions, especially due to user mobility, channel fading, and interference conditions, the performance of TCP degrades. The root of this degradation rests in the difficulty for TCP to distinguish between congestion, contention, and channel errors. Moreover, the wireless MAC may cause unfairness for the transport layer congestion control:

when more nodes contend for access to the wireless resource, the node that first wins the contention achieves a better capacity (i.e., higher congestion window value). Finally, the standard TCP congestion control [5] [7] mechanism is known to perform poorly over satellite broadband links due to both the large Round-Trip Time (RTT) value and the typically high packet error rates.

TCP Variants are

- **TCP SACK**

In TCP SACK cumulative acknowledgement reside in acknowledgment number field. Cumulative acknowledgement field is used to indicating all of the data up to the marked byte that is received by TCP receiver. A selective acknowledgement option allows receivers to additionally report up to three non sequential blocks of data they have successfully received. So the sender does only need to retransmit the missing TCP segments [3] [4].

- **TCP Tahoe**

An early version of TCP included slow start congestion avoidance and fast retransmits mechanism to control congestion in networks. Further development in the original TCP version includes a modification to the RTT estimator used to evaluate retransmission timeout values. The problem of TCP Tahoe [3] [4] is that if the error was random in nature then it is not always efficient. In such a case the massive shrinkage of the congestion window is unnecessary. In such a situation TCP is not capable to fully consume the available bandwidth of the radio channel during the phase of window re-expansion.

- **TCP RENO**

The TCP Reno [4] implementation uses the enhancements integrated into Tahoe, but introduced fast recovery in conjunction with fast retransmit. Using fast recovery significantly improves performance compared to TCP Tahoe when a single packet is lost from a window of data, but can negatively impact performance for multiple packets drop from a single window of data.

TCP NEW RENO

TCP New Reno [3] includes a small change to the TCP Reno algorithm at the source side that eliminates the waiting time for a retransmit timer to expire when number of packets are lost from a window. When a partial acknowledgement is received that acknowledges some received packets, but not acknowledge all of the packets that were outstanding then at the start of that fast recovery procedure the change incorporate the sender's behavior during fast recovery.

SIMULATION MODEL

OPNET (14.5) simulator is used for deploying UMTS network architecture by means of different nodes (mobile & fixed) from object palette. OPNET MODELER is used for design and study communication networks, devices, protocols and applications. OPNET provides a graphical user interface to build simulation models for various network entities from physical layer modulator to application processes [8] [9].

- **Simulation Scenario**

For implementation of various TCP variants four scenarios have been created, i. e TCP-Tahoe, TCP SACK, TCP

RENO and TCP NEWRENO. A single scenario completed in all aspects, duplicated and then attributes are set for both the scenarios. Each scenario is employed only for file transfer by using Background class. Each scenario is designed for five users with their movement across Node-B. Along with users, simulation model consists of following entities: one Node-B access points, RNC, SGSN, GGSN and one FTP servers for single type of traffic class. For connectivity between nodes various links were used form object palette. After the architecture is completed, the required attributes are set for each node. Applications are defined in the application configuration node and packet discarder utility is used to discard the packet at particular time interval. For implementation of each TCP variants four different scenarios are designed for measuring the different global and object statistics. Figure 2 shows the simulation scenario

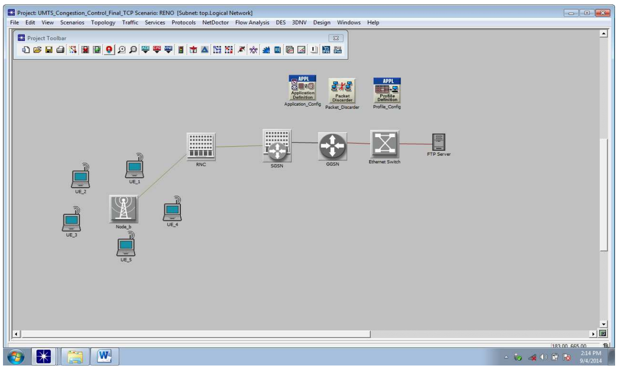

Figure 2: TCP Simulation Scenario

- **Simulation Parameters**

For analysis of results various parameters needs to be considered are Scenario parameters, profile configuration parameters and Packet Discarder parameters.

- **Scenario Parameters*:*** For each scenario certain parameters are considered and needs to be set as shown in table 1.

Table 1: Simulation Parameter

Simulation Parameters	Value
Simulation Time	600 Sec
Number of Nodes	05
Environment Size	Logical Environment
Traffic Type	Constant Bit Rate
Seed	300
Value per Statistics	300
Update Interval	500000
Simulation	Based on Kernel type Preferences
Number of runs	One for each scenario

SIMULATION RESULTS

- **Global Statics**

Figure 3 shows download response time on FTP server for different variants of TCP. Download response time for TCP

NEW RENO is much good rather than other TCP variants.

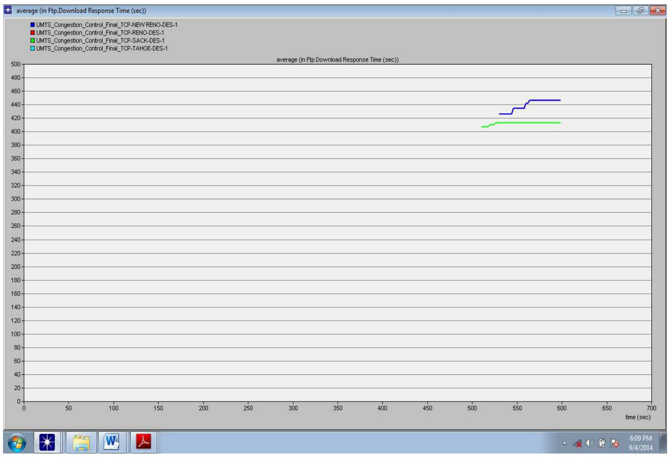

Figure 3: FTP Download Response Time

Figure 4 shows the Traffic Received and Send at FTP Server for different TCP variants and TCP NEW RENO shows better result for FTP traffic in UMTS networks.

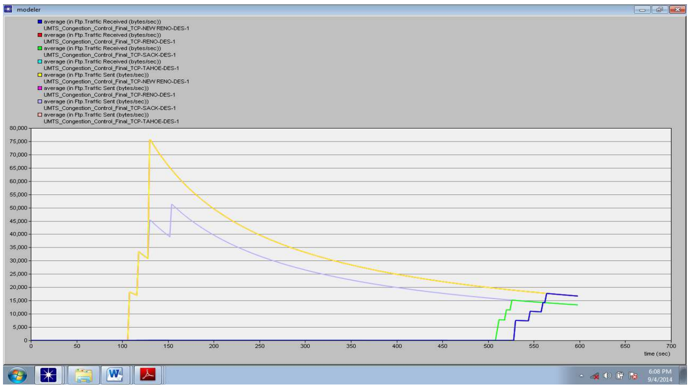

Figure 4: Traffic Received and Send at FTP Server

- **Object Statistics**

The number of statistics is collected on various nodes in the UMTs network. Node statistics is collected for variants of TCP TCP Tahoe, TCP RENO, TCP SACK and TCP NEW RENO. Figure 7 graph shows the TCP congestion window on FTP server for TCP variants.

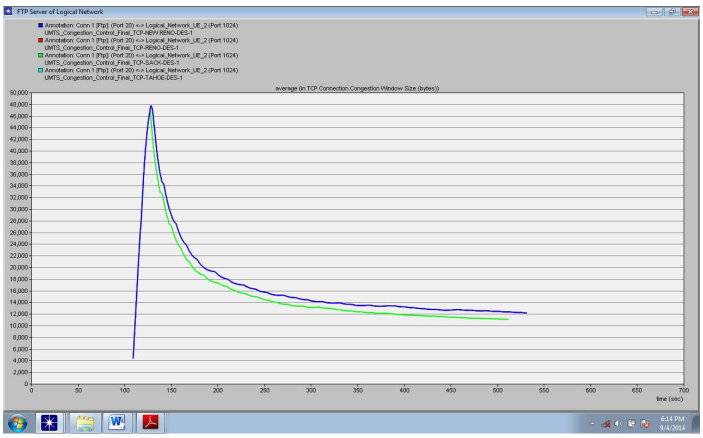

Figure 5: TCP Congestion Window Size

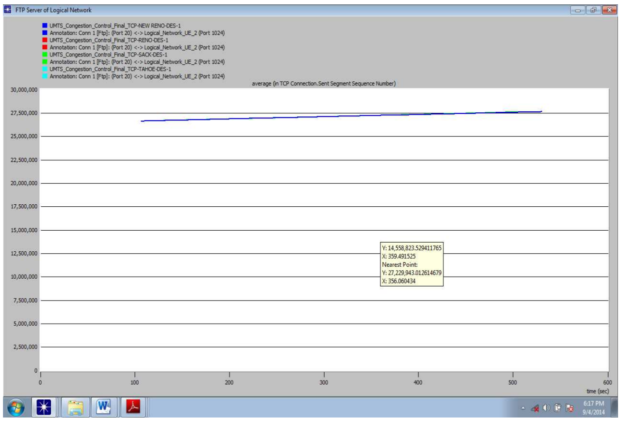

Figure 6: TCP Connection Sent Segment Sequence Number

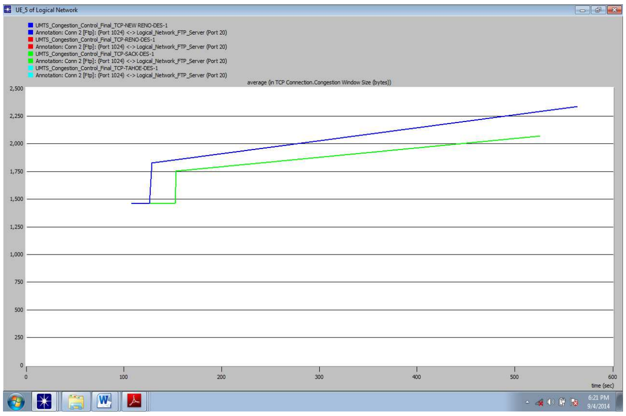

Figure 7: TCP Congestion Window at UE

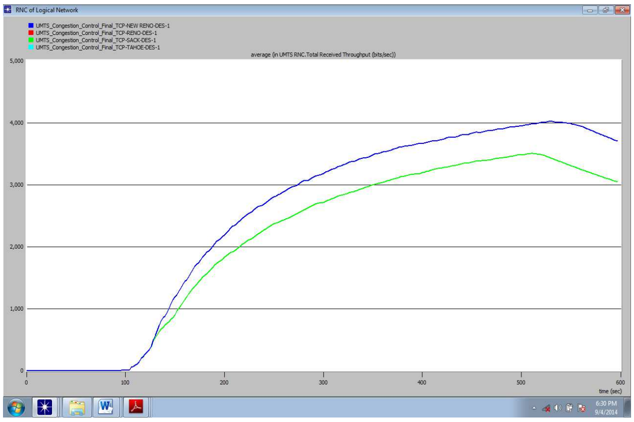

Figure 8: Received Throughputs at RNC

CONCLUSIONS AND FUTURE WORK

This paper explores out the conception of TCP variant over the UMTS network using OPNET modeler 14.5 simulator. For assessment of congestion window for different TCP variants, scenarios for all TCP variants i.e. TCP Tahoe, TCP SACK, TCP RENO and TCP New RENO are formedand assorted parameter values are delineated. By investigation assortment of statistics of handover it is winded up that TCP New RENO provides better performance as compared to other TCP variants on wireless environment. In the near future more and more services running over TCP will be offered in high-speed wired and wireless environments. However, the new high-speed environments like UMTS exceed the range for

which TCP was initially designed, tested and tuned. TCP shows some undesirable patterns of behavior in the context of wireless networks. As a consequence more active research in the field of TCP over wireless networks (WLAN, UMTS) is in progress to modify the protocol to to reduce the packet losses on wireless networks.

ACKNOWLEDGEMENTS

Authors are greatful to Gyan Vihar University Jaipur for providing support to carried out this research work**.**

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